Grandstream Networks, Inc.

Analog IP Gateway GXW42XXseries
16, 24, 32 or 48FXS Ports

www.lapdattongdai.com

GXW42XXUSER MANUAL INDEX

GNU GPL INFORMATION 5

CHANGE LOG 6

CHANGES FROM 1.0.4.9 USER MANUAL 6

CHANGES FROM 1.0.4.4 USER MANUAL 6

CHANGES FROM 1.0.3.9 USER MANUAL 6

WELCOME 7

GATEWAY GXW42XXOVERVIEW 7

SAFETY COMPLIANCE 7

WARRANTY 8

CONNECT YOUR GXW42XXGATEWAY 9

EQUIPMENT PACKAGING 9

CONNECT THE GXW42XX 9

GXW42XXFEATURES 14

SOFTWARE FEATURES OVERVIEW 14
HARDWARE SPECIFICATION 17
GXW42XX LCD MENU 18
BASIC OPERATIONS 19
UNDERSTANDING GXW VOICE PROMPTS 19
PLACING A PHONE CALL 20
PHONE OR EXTENSION NUMBERS 20
DIRECT IP CALLS 21
CALL HOLD 22
CALL WAITING 22
CALL TRANSFER 22
BLIND TRANSFER 22
ATTENDED TRANSFER 23
3-WAY CONFERENCING 23
HUNTING GROUP 23
INTER-PORT CALLING 25
SENDING AND RECEIVING FAX 26
SUPPORT RADIUS PROTOCOL 26
CALL FEATURES 27
CONFIGURATION GUIDE 29
CONFIGURING GXW42XX VIA VOICE PROMPT 29
CONFIGURING GXW42XX WITH WEB BROWSER 29
ACCESS THE WEB CONFIGURATION MENU 30
IMPORTANT SETTINGS 30
NAT SETTINGS 30
DTMF METHODS 31
PREFERRED VOCODER (CODEC) 31
SAVING THE CONFIGURATION CHANGES 50
REBOOTING FROM REMOTE 51
CONFIGURATION THROUGH A CENTRAL SERVER 51
SOFTWARE UPGRADE 52
FIRMWARE UPGRADE THROUGH TFTP/HTTP/HTTPS 52
INSTRUCTIONS FOR LOCAL FIRMWARE UPGRADE 53
CONFIGURATION FILE DOWNLOAD 53
FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX 54
MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD 54
RESTORE FACTORY DEFAULT SETTING 55

www.tapdatongdaidienthoai.com

FACTORY RESET 55
RESET BUTTON *55*
IVR COMMAND *55*

www.lapdattongdaidienthoai.com

TABLE OF FIGURES

GXW42XX User Manual

Figure 1: Diagram of GXW4216/24/32/48 Panel 11
Figure 2: 50 Pin Telco Connector 12
Figure 3: LCD Menu 18

TABLE OF TABLES

GXW42XX User Manual

Table 1: Definitions of the GXW Connectors 11
Table 2: Definitions of the GXW Display Panel 11
Table 3: GXW42XX Software Features 14
Table 4: Hardware Specification of GXW42XX series gateways 17
Table 5: Definitions of the GXW Voice Prompts 19
Table 6: Call Features Table (Star Code) 27
Table 7: Status 32
Table 8: Maintenance 33
Table 9: Advanced settings 38
Table 10: Profiles 39
Table 11: FXS Ports 50

GNU GPL INFORMATION

GXW42xx firmware contains third-party software licensed under the GNU General Public License (GPL). Grandstream uses software under the specific terms of the GPL. Please see the GNU General Public License (GPL) for the exact terms and conditions of the license.

Grandstream GNU GPL related source code can be downloaded from Grandstream web site from: http://www.grandstream.com/support/faq/gnu_gpl .

CHANGE LOG

This section documents significant changes from previous versions of GXW42xx user manuals. Only major new features or major document updates are listed here. Minor updates for corrections or editing are not documented here.

CHANGES FROM 1.0.4.9 USER MANUAL

- Added option [**Use Request Routing ID in SIP Headers**] to enable/disable device to use configured Request URI Routing ID for certain FXS port in its SIP message Header when a trunk SIP account is used.
- Added option [**Add Auth Header on Initial REGISTER**] to enable/disable including authentication header in the initial REGISTER.

CHANGES FROM 1.0.4.4 USER MANUAL

- Added option [**Enable LLDP**] to enable/disable LLDP.
- Added field [**Request URI Routing ID**] to allow device to route the calls to individual fxs ports based on the DID, when only have one sip registration.

CHANGES FROM 1.0.3.9 USER MANUAL

www.lapdattongdaidienthoai.com

- Added option [**Display Language**] to choose language for LCD display.
- Added option [**Prompt Dial Tone Code**] to configure dial code.

WELCOME

Thank you for purchasing the Grandstream GXW42XXAnalog FXSIP Gateway. The GXW42XX offers an easy to manage, easy to configure IP communications solution for any business with virtual and/or branch locations. The GXW42XX supports popular voice codecs and is designed for full SIP compatibility and interoperability with third party SIP providers, thus enabling you to fully leverage the benefits of VoIP technology, integrate a traditional phone system into a VoIP network, and efficiently manage communication costs.

This manual will help you learn how to operate and manage your GXW FXS Analog IP Gateway and make the best use of its many upgraded features including simple and quick installation, multi-party conferencing, and direct IP-IP Calling. This IP Analog Gateway is very easy to manage and scalable, specifically designed to be an easy to use and affordable VoIP solution for the small – medium business or enterprise.

GATEWAY GXW42XX OVERVIEW

The new GXW42XXseries has a compact and quiet design and offers superb audio quality, rich feature functionality, strong security protection, and good manageability. It is auto-configurable, remotely manageable and scalable.

The GXW42XXseriesfeatures 16, 24, 32 or 48port FXS interface for analog telephones, dual 10/100/1000Mbps network ports, and RJ21analog port. In addition, it supports the option of 4 SIP Server profiles, caller ID for various countries/regions, T.38 fax, flexible dialing plans, security protection (SIPS/TLS), comprehensive voice codec including G.711 (a/u-law), G.723.1, G.726(16/24/32/40 bit rates),iLBC and G.729.

SAFETY COMPLIANCE

The GXW42XXis compliant with various safety standards including FCC/CE. Its power adapter is compliant with UL standard. **Warning:** use only the power adapter included in the GXW42XX package. Use of alternative power adapter may permanently damage the unit.

WARRANTY

Grandstream has a reseller agreement with our reseller customers. End users should contact the company from whom the product was purchased, for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Support for an RMA (Return Materials Authorization) number. Grandstream reserves the right to change thewarranty policy without prior notification.

Caution: *Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.*

This document contains links to HT502 GUI Interfaces. Please download these examples from

www.lapdattongdaidienthoai.com

http://www.grandstream.com/products/gxw_series/gxw42xx/documents/gxw42xx_gui.zip for your reference. This document is subject to change without notice. The latest electronic version of this user manual is available for download at:

http://www.grandstream.com/products/gxw_series/gxw42xx/documents/gxw42xx_usermanual_english.pdf

Reproduction or transmittal of the entire or any part, in any form or by any means, electronic or print, for any purpose is not permitted without the express written permission of Grandstream Networks, Inc.

CONNECT YOUR GXW42XXGATEWAY

Connecting the GXW42XX gateway is easy. Before you begin, please verify the contents of the GXW42XXpackage.

EQUIPMENT PACKAGING

Unpack and check all accessories. Equipment includes:

- one device unit
- one RJ45 Ethernet cable
- one 12V 5A universal power adapter (24V 6.25A for GXW4248)
- mount

CONNECT THE GXW42XX

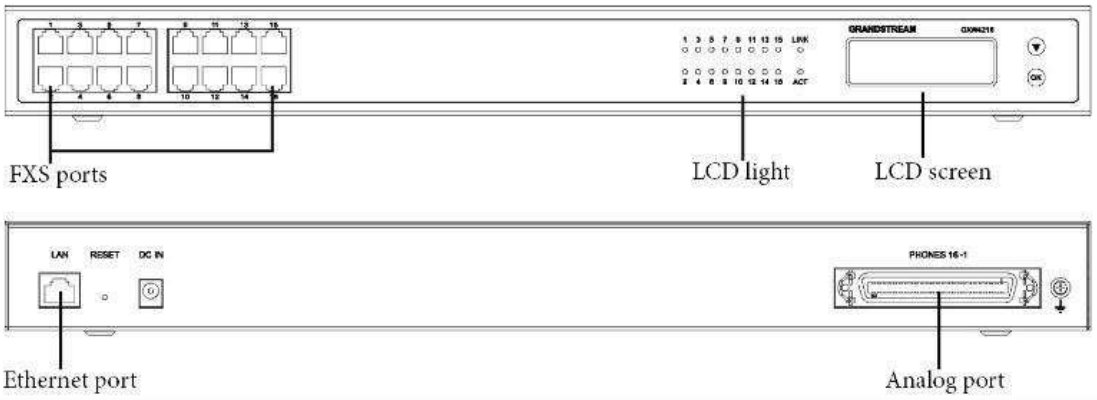
Follow these four (4) steps to connect your GXW42XXgateway to the Internet and access the unit's configuration pages.

1. Connect standard touch-tone analog phones to the GXW42XX's RJ21 port with a RJ11 to R21 cable.
2. Insert anRJ45 Ethernet cable into the WAN port of GXW42XX and connect the other end to an uplink port (a router or a modem, etc.)
3. Plugthe power adapter into the GXW42XX gateway into a poweroutlet.

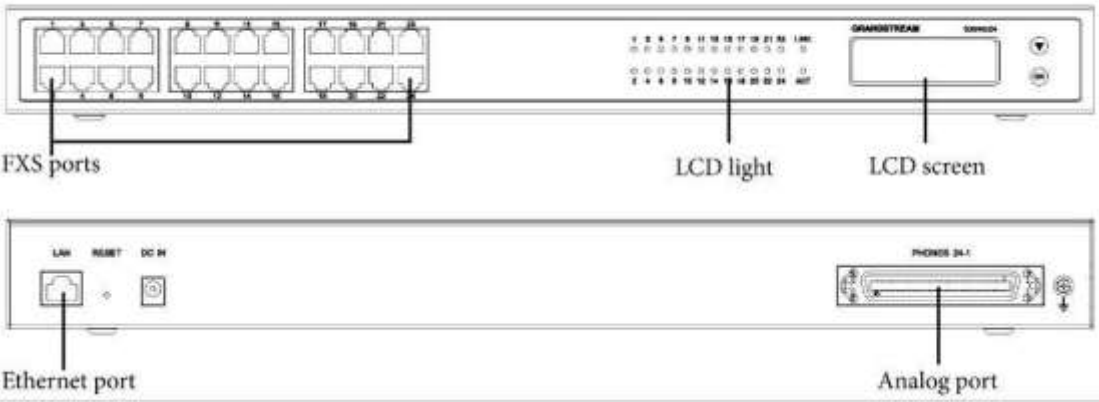
Follow the instructions from the topic "Configuring GXW 42XX with Web Browser" for initial configuration. The GUI pages will guide you through the remaining steps to set-up your gateway.

www.laptoptongdaidienthoai.com

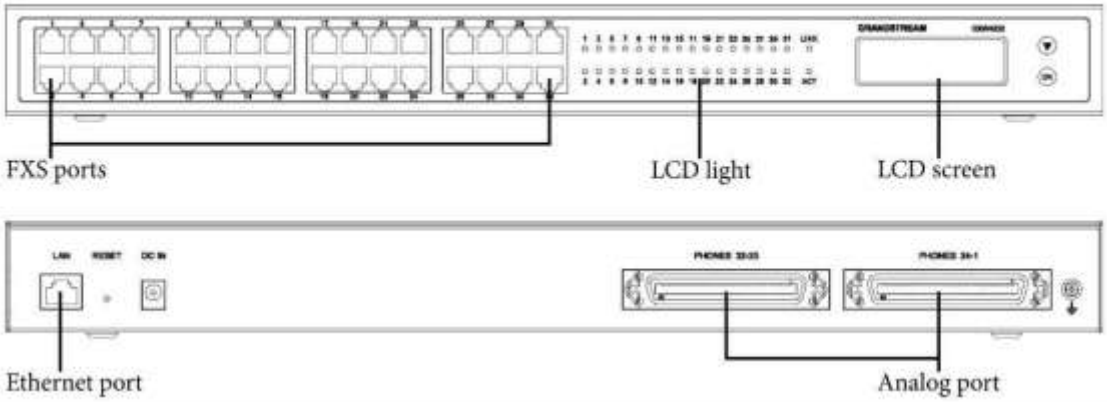
GXW4216



GXW4224



GXW4232



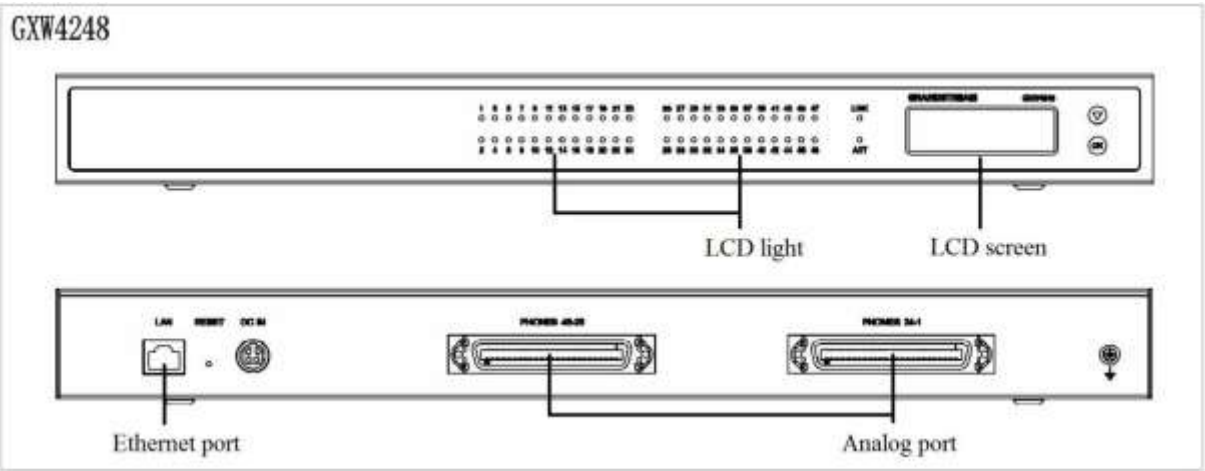


FIGURE 1: DIAGRAM OF GXW4216/24/32/48 PANEL

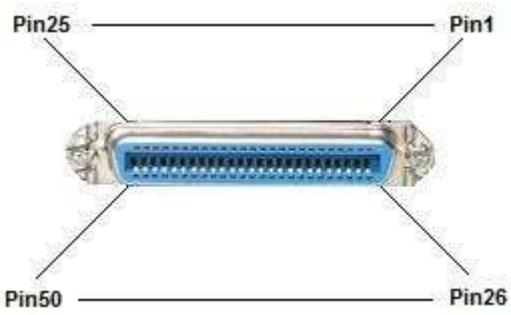
TABLE 1: DEFINITIONS OF THE GXW CONNECTORS

Ethernet port	Connect to the internal LAN network or router
RESET	Factory Reset button. Press and hold for a while to reset factory default settings.
DC IN	Power adapter connection
Analog port	Connect to analog phones / fax machines with an RJ21 to RJ11 cable
FXS ports	FXS port to be connected to analog phones / fax machines.

Note: Once the GXW42XX is turned on and configured, the front display panel indicates the status of the unit.

TABLE 2: DEFINITIONS OF THE GXW DISPLAY PANEL

Act LED	Remains ON if plug the network cable.
LINK LED	Indicates Ethernet port activity.
FXS LED	Indicate status of the respective FXS Ports on the back panel **Busy** - ON (Solid Green) **Available** - OFF Slow blinking FXS LEDs indicates Voice Mail for that port. All FXS LEDs slow blinking indicates provisioning.



Pin	Signal	Pin	Signal
1	RING1	26	TIP1
2	RING2	27	TIP2
3	RING3	28	TIP3
4	RING4	29	TIP4
5	RING5	30	TIP5
6	RING6	31	TIP6
7	RING7	32	TIP7
8	RING8	33	TIP8
9	RING9	34	TIP9
10	RING10	35	TIP10
11	RING11	36	TIP11
12	RING12	37	TIP12
13	RING13	38	TIP13
14	RING14	39	TIP14
15	RING15	40	TIP15
16	RING16	41	TIP16
17	RING17	42	TIP17
18	RING18	43	TIP18
19	RING19	44	TIP19
20	RING20	45	TIP20
21	RING21	46	TIP21
22	RING22	47	TIP22
23	RING23	48	TIP23
24	RING24	49	TIP24

FIGURE 2: 50 PIN TELCO CONNECTOR

NOTE:

- For GXW4216, only first 16 pairs are connected.
- For GXW4232, first 24 pairs on 1st RJ21 connector and first 8 pairs on the 2nd RJ21 connector are connected.
- For GXW4224 and GXW4248, the last pair of pin 25 and 50 on each RJ21 is not connected.
- All LED lights display green when ON.
- LINK LED and ACT LED blinking in following status indicate different network speed.

  10M: LINK and ACT all blink, ACT blinks faster then LINK

  100M: LINK and ACT all blink, ACT blinks the same fast as LINK

  1000M: LINK does not blink, only ACT blink
- Slow blinking of FXS LED together indicates a firmware upgrade or provisioning state.

GXW42XXFEATURES

The GXW42XXis a next generation IP voice gateway that is interoperable and compatible with leading IP-PBXs, Softswitches and SIP platforms. The GXW42XXFXS gateway is auto-configurable, remotely manageable and scalable. The GXW42XX gateways come in fourmodels - the GXW4216, GXW4224, GXW4232and GXW4248,each offering superb voice quality, traditional telephony functionality, easy deployment, and 16, 24,32 and 48FXS ports respectively.Each model features flexible dialing plans, integrated call routing to support a pure IP network call and an external power supply.

SOFTWARE FEATURES OVERVIEW

- 16, 24, 32 or 48FXS ports (no front panel FXS ports on GXW4248)
- RJ-45 Ethernet ports
- 4 configurable SIP profiles
- Supports Voice Codecs:

  G711(a/μ, Annex I & II), G723.1A, G726 (ADPCM with 16/24/32/40 bit rates), G729 A/B,iLBC,T.38 Fax
- Comprehensive Dial Plan support for Outgoing calls.
- G.168 Echo Cancellation
- Voice Activation Detection (VAD), Comfort Noise Generation (CNG), and Packet Loss Concealment (PLC)
- Supports PSTN/PBX analog telephone sets or analog trunks

TABLE 3: GXW42XXSOFTWARE FEATURES

	GXW4216	GXW4224	GXW4232	GXW4248
Telephone Interfaces	16 FXS ports	24 FXS ports	32 FXS ports	48 FXS ports
SIP Provisioning	16 SIP accounts, 4 profiles	24 SIP accounts,4 profiles	32 SIP accounts, 4 profiles	48 SIP accounts, 4 profiles
Network Interface	10/100/1000 Mbps, RJ-45	10/100/1000 Mbps, RJ-45	10/100/1000 Mbps, RJ-45	10/100/1000 Mbps, RJ-45
Number of Concurrent Calls (except when using SRTP)	16 Concurrent Calls	24 Concurrent Calls	32 Concurrent Calls	48 Concurrent Calls

ngdaidienthoai.com

Voice over Packet Capabilities	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), LEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), LEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), LEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711	Voice Activity Detection (VAD) with CNG (comfort noise generation) and PLC (packet loss concealment), LEC with NLP Packetized Voice Protocol Unit (supports RTP and AAL2 protocol), G.168 compliant Echo Cancellation, Dynamic Jitter Buffer, Modem detection & auto-switch to G.711
Voice Compression	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.722 (pending), G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729, iLBC, G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.722 (pending), G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729, iLBC, G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.722 (pending), G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729, iLBC, G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet	G.711 + Annex I (PLC), Annex II (VAD/CNG format) encoder and decoder, G.722 (pending), G.723.1A, G.726(ADPCM with 16/24/32/40 bit rates), G.729, iLBC, G.726 provides proprietary VAD, CNG, and signal power estimation, Voice Play Out unit (reordering, fixed and adaptive jitter buffer, clock synchronization), AGC (automatic gain control), Status output, Decoder controlling via voice packet

	header	header	header	header
DHCP Server/Client	DHCP Client only	DHCP Client only	DHCP Client only	DHCP Client only
Fax over IP	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.21, V.27ter, V.29 for T.38 fax relay	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.21, V.27ter, V.29 for T.38 fax relay	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.21, V.27ter, V.29 for T.38 fax relay	T.38 compliant Group 3 Fax Relay up to 14.4kpbs and auto-switch to G.711 for Fax Pass-through, Fax Datapump V.17, V.21, V.27ter, V.29 for T.38 fax relay
QoS	DiffServ, TOS, 802.1P/Q VLAN tagging	DiffServ, TOS, 802.1P/Q VLAN tagging	DiffServ, TOS, 802.1P/Q VLAN tagging	DiffServ, TOS, 802.1P/Q VLAN tagging
Transport Protocol	RTP	RTP	RTP	RTP
DTMF Method	In-audio, RFC2833, and/or SIP Info	In-audio, RFC2833, and/or SIP Info	In-audio, RFC2833, and/or SIP Info	In-audio, RFC2833, and/or SIP Info
IP Signaling	SIP (RFC 3261)	SIP (RFC 3261)	SIP (RFC 3261)	SIP (RFC 3261)
Provisioning	TFTP, HTTP, HTTPS	TFTP, HTTP, HTTPS	TFTP, HTTP, HTTPS	TFTP, HTTP, HTTPS
Security	SRTP, IPSEC, TLS/SIPS, HTTPS, 802.1x	SRTP, IPSEC, TLS/SIPS, HTTPS, 802.1x	SRTP, IPSEC, TLS/SIPS, HTTPS, 802.1x	SRTP, IPSEC, TLS/SIPS, HTTPS, 802.1x
Management	Syslog support, HTTP and Telnet access	Syslog support, HTTP and Telnet access	Syslog support, HTTP and Telnet access	Syslog support, HTTP and Telnet access
Dial Plan	Yes	Yes	Yes	Yes
3-Way Conference	3-Way conference with local mixing	3-Way conference with local mixing	3-Way conference with local mixing	3-Way conference with local mixing
Caller ID	Bellcore Type 1 & 2, ETSI, BT, NTT, FSK and DTMF-based CID	Bellcore Type 1 & 2, ETSI, BT, NTT, FSK and DTMF-based CID	Bellcore Type 1 & 2, ETSI, BT, NTT, FSK and DTMF-based CID	Bellcore Type 1 & 2, ETSI, BT, NTT, FSK and DTMF-based CID
Polarity Reversal / Wink	Yes	Yes	Yes	Yes
Network Connectivity	IPv4, IPv6, TCP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS,DHCP, NTP,	IPv4, IPv6, TCP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS,DHCP, NTP,	IPv4, IPv6, TCP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS,DHCP, NTP,	IPv4, IPv6, TCP/UDP, RTP, HTTP/HTTPS, ARP/RARP, ICMP, DNS,DHCP, NTP,

	TFTP, TELNET, PPPoE, STUN	TFTP, TELNET, PPPoE, STUN	TFTP, TELNET, PPPoE, STUN	TFTP, TELNET, PPPoE, STUN

HARDWARE SPECIFICATION

The hardware specifications of the GXW FXS series are detailed in Table 4.

TABLE 4: HARDWARE SPECIFICATION OF GXW42XX SERIES GATEWAYS

	GXW4216	GXW4224	GXW4232	GXW4248
Telephone Interface	16 RJ11 Ports/ 1 RJ21 Port	24 RJ11 Ports/ 1 RJ21 Port	32 RJ11 Ports/ 2 RJ21 Ports	2 RJ21 Ports only
FXS LEDs	16	24	32	48
Network interface	LAN, Single 10/100/1000 BASE-TX,RJ45			
Power Input	**Input**:100-240VAC, 50/60Hz **Output**: 12V DC, 5.0A(4216/24/32) 24V DC, 6.25A(4248 only)			
LCD screen	128×32 pixel			
Telco connector	1 RJ21 (50 pins)	1 RJ21 (50 pins)	2 RJ21 (50 pins)	2 RJ21 (50 pins)
RJ-11 connectors	Yes	Yes	Yes	No
NAND Flash	64MB			
DRAM	128MB (+ separate 8MB if with a slave processor)			
Function Buttons	1 button for Reset/Factory Reset			
Environmental	Operation: 0°C to 45°C Storage: -20°C to 60°C Humidity: 10% to 90% Non-condensing			
Mounting	Desktop and Rack mount			
Onhook Voltage	Fixed, 48V			
Ring Voltage	50Vrms (balanced ringing)			
Ring Frequency	20-50Hz			
Short Haul Loop	2REN: Up to 2km on 24 AWG wire	2REN: Up to 2km on 24 AWG wire	2REN: Up to 2km on 24 AWG wire	2REN: Up to 2km on 24 AWG wire
Outdoor Protection	Over-voltage Protection and surge immunity			
Signaling	FXS Loop-start			

EMC	EN55022/EN55024 and FCC part15 Class B
Safety	UL
Compliance	FCC, CE, C-Tick

GXW42XX LCD MENU

The GXW42XX gateway series includes a small LCD screen for the display of basic information. The LCD has a display area of 128x32 pixels, which will allow for 2 lines of text with a 16px height limitation per line. The LCD menu is showed as Figure 3: LCD Menu. Menu is navigated by the Down-arrow and OK button.

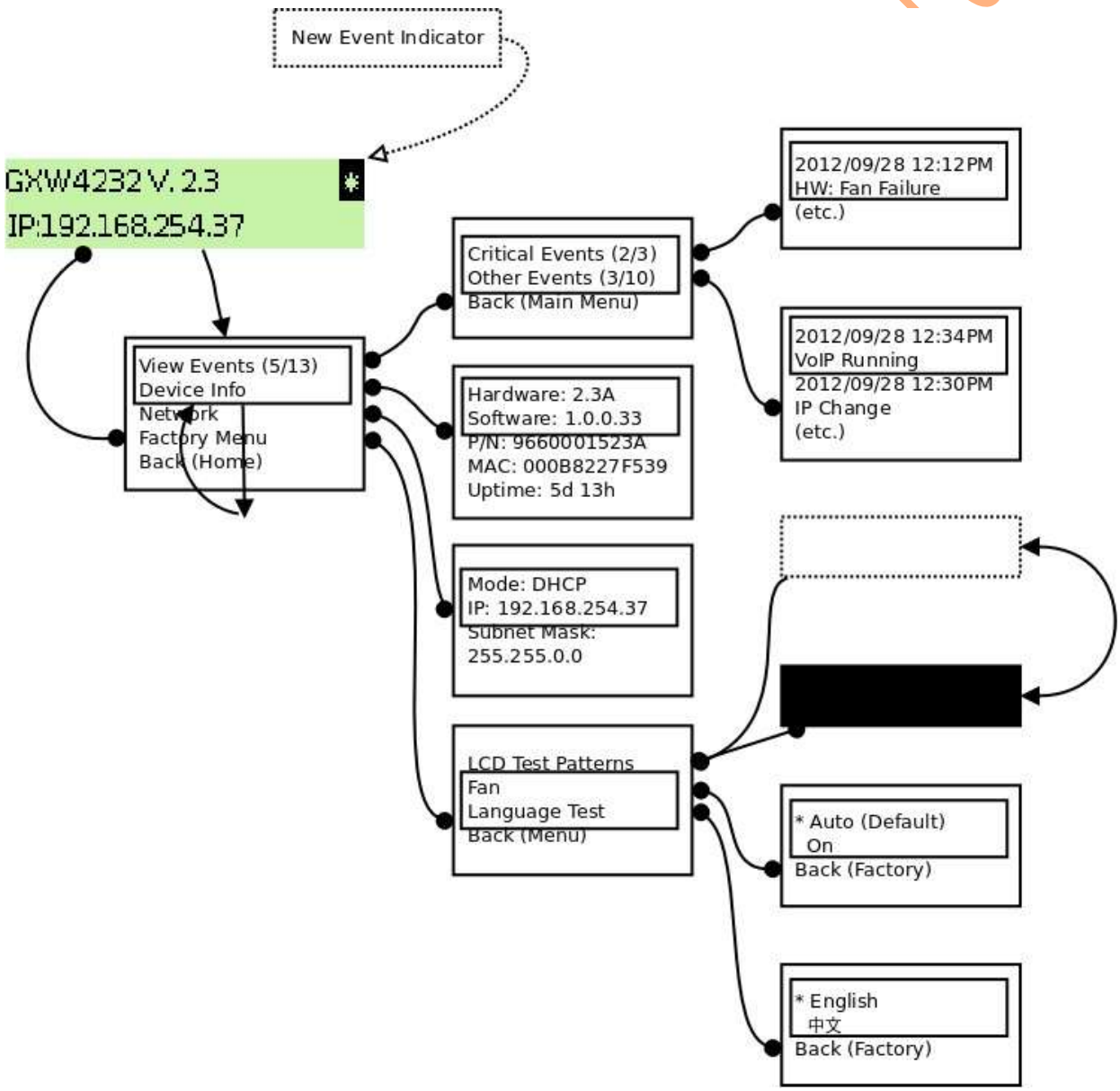


FIGURE 3: LCD MENU

BASIC OPERATIONS

UNDERSTANDING GXW VOICE PROMPTS

GXW42XX has a built-in voice prompt menu for simple device configuration. To enter the voice prompt menu, press ******* on the standard analog phone which is connected to device's FXS port.

TABLE 5: DEFINITIONS OF THE GXW VOICE PROMPTS

Menu	Voice Prompt	User's Options
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter "001 – 005,007, 010, 013 - 017,047, 086 or 099" Menu option Enter "9" for confirming a option
001	"DHCP Mode", "PPPoEMode "or"Static IP Mode"	Enter '9' to toggle the selection *If user selects "Static IP Mode", user need configure all the IP address information through menu 002 to 005. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.*
002	"IP Address " + IP address	The current WAN IP address is announced Enter 12-digit new IP address if in Static IP Mode.
003	"Subnet " + IP address	Same as Menu option 002
004	"Gateway " + IP address	Same as Menu option 002
005	"DNS Server " + IP address	Same as Menu option 002
007	Preferred Vocoder	Enter "9" to go to the next selection in the list: • PCM U • PCM A • iLBC • G-726 • G-723 • G-729
010	"MAC Address"	Announces the Mac address of the unit.
013	Firmware Server IP Address	Announces current Firmware Server IP address. Enter 12 digit new IP address.
014	Configuration Server IP Address	Announces current Config Server Path IP address. Enter 12 digit new IP address.
015	Upgrade Protocol	Upgrade protocol for firmware and configuration update. Enter

		"9" to toggle between **TFTP** and **HTTP**
016	Firmware Version	Firmware version information.
017	Firmware Upgrade	Firmware upgrade mode. Enter "9" to rotate among the following three options: 1. always check 2. check when pre/suffix changes 3. never upgrade
047	"Direct IP Calling"	Enter the target IP address to make a direct IP call, after dial tone. (See "*Make a Direct IP Call*".)
086	Voice Mail	Number of voice mails
099	"RESET"	Enter "9" to reboot the device; or Enter MAC address to restore factory default setting (See **Restore Factory Default Setting** section)
700-748	Phone calls between different ports of the same GW42xx	GXW42xX support inter-port calling from voice menu for easy test/verification in factory "700" Ring all ports and connect to first port pick up the call "701-732" Call individual port
	"Invalid Entry"	Automatically returns to Main Menu

Five Success Tips when using the Voice Prompt

1. "*" shifts down to the next menu option
2. "#" returns to the main menu
3. "9" functions as the ENTER key in many cases to confirm an option
4. All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. For IP address, add **0** before the digits if the digits are less than 3 (i.e. - 192.168.0.26 should be key in like 192168**000**026. No decimal is needed).
5. Key entry cannot be deleted but the phone may prompt error once it is detected

PLACING A PHONE CALL

PHONE OR EXTENSION NUMBERS

1. Dial the number directly and wait for 4 seconds (To change the default value, modify the following setting - "No Key Entry Timeout")

or

2. Dial the number directly and press # (Use # as dial key" must be configured in web configuration).

Examples:

1. Dial a number (e.g. (626) 666-7890), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 4 seconds. Check with your VoIP service provider for further details on prefix numbers.

DIRECT IP CALLS

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to

each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both GXW42XXand other VoIP Device, have public IP addresses, or
2. Both GXW42XXand other VoIP Device are on the same LAN using private IP addresses, or
3. Both GXW42XXand other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

GXW42XXsupports two ways to make Direct IP Calling:

Using IVR

1. Pick up the analog phone then access the voice menu prompt by dial "***"
2. Dial "**047**" to access the direct IP call menu
3. Enter the IP address using format ex. **192*168*0*160** after the dial tone.

Using Star Code

1. Pick up the analog phone then dial "***47**"
2. Enter the target IP address using same format as above. Note:

NO dial tone will be played between step 1 and 2.

Destination ports can be specified by using "*****" (encoding for ":") followed by the port number.

Examples:

a) If the target IP address is 192.168.0.160, the dialing convention is ***47 or Voice Prompt with option 047, then 192*168*0*160.** followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

b) If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be: ***47 or Voice Prompt with option 047, then 192*168*0*160*5062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

CALL HOLD

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button). Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

CALL WAITING

Call waiting tone (2 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

CALL TRANSFER

BLIND TRANSFER

Assume that call Caller A and B are in conversation. A wants to *Blind Transfer* B to C:

3. Caller A presses **FLASH** on the analog phone to hear the dial tone.
4. Caller A dials ***87** then dials caller C's number, and then # (or wait for 4 seconds).
5. Caller A will hear the confirm tone. Then, A can hang up.

NOTE: "*Enable Call Feature*" must be set to "Yes" in web configuration page. Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial tone. This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, Caller A can either hang up or make another call.

www.landattongdaidienthoai.com

2. A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
3. Continuous busy tone. The phone has timed out. **Note:** continuous busy tone does not indicate the transfer has been successful, nor does it indicate the transfer has failed. It often means there was a failure to receive second NOTIFY – check firmware for most recent release.

ATTENDED TRANSFER

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.
2. Caller A then dials Caller C's number followed by # (or wait for 4 seconds).
3. If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer.
4. If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

NOTE: When Attended Transfer fails and A hangs up, the GXW42XXwill ring back user A to remind A that B is still on the call. A can pick up the phone to resume conversation with B.

3-WAY CONFERENCING

The GXW42XXsupports Bellcore style 3-way Conference.

Instructions for 3-way conference:

Assuming that call party A and B are in conversation. A (GXW42XX) wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C's number then # (or wait for 4 seconds).
3. If C answers the call, then A presses FLASH to bring B, C in the conference.
4. If C does not answer the call, A can press FLASH back to talk to B.
5. If A presses FLASH during conference, C will be dropped out.
6. If A hangs up, the conference will be terminated or transfer B to C if "Transfer on Conference Hangup" set to yes

HUNTING GROUP

This feature allows the user to setup a single SIP account on the gateway and have the ability to use all FXS ports to make/receive calls. Using this feature, all ports active in same Hunting Group will have the same phone number and incoming calls will be distributed in a Linear or Circular manner among the ports active in that Hunting Group. The number of hunting groups is limited by the number of ports each GXW model has - i.e. each port can be its own Hunting Group. The most practical and efficient way to use

Hunting Groups is to assign 2 or 3 ports to separate Hunting Groups.

One additional and popular way to use the Hunting Group feature is called "*multiplexed analog lines*". In this configuration, a legacy PBX system with 8 FXO trunks can be connected to 8 GXW 42xxports configured as a Hunting Group. The GXW can be registered to a SIP server provider using only <u>one</u> phone number. If the SIP service provider allows multiple calls to the same number, the GXW will allow 8 concurrent calls to the same SIP number. All office members can be reached remotely using the samephonenumber in a round-robin fashion.

<u>Example Configuration of a typical Hunting Group</u>:

1. Configure the SIP account from your VoIP Service Provider on **FXS port 1**under**FXS Ports**webpage.
2. Select **Active** under the **Hunting Group** drop box for FXS port 1.

www.lapdattongdaidienthoai.com

3. For the remaining ports (say 2, 3 and 4) select **1** for **Hunting Group**.Ports 2, 3 and 4 are now active members of the hunting group associated with port 1.

This configuration will route all calls directed to FXS port 1 to ports 2, 3 and/or 4 in round robin fashion respectively *if* port 1 is busy or times out. You can configure the ring timeout on the **Profile** page.

Example configuration of a multipleHunting Group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #2: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**" FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**" FXS Port #4: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**" FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**4**" FXS Port #7: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**7**"

…

Hunting Group 1 contains ports 1, 2, 3. Hunting Group 4 contains ports 4, 5, 6.

Hunting Group 7 contains ports 7, 8.

Please be aware, the choice of 1 for ports 2 and 3, the choice of 4 for ports 5 and 6, the choice 7 for port 8 is required to indicate that the SIP account tied to port marked as **"Active"** will be used for all members of the same Hunting group. Needless to say, those members of the same Hunting group may not be sequential ports. In following example ports 3, 5 and 7 tied to SIP Account configured in Port #1 marked as **"Active"**, and ports 4,6,8 tied to SIP Account configured in Port #2 marked as **"Active"** as well.

Example of not sequential configuration of a multipleHunting Group:

FXS Port #1: SIP UserID and Authenticate ID entered, Hunting group set to "**Active**" FXS Port #2: SIP UserID and Authenticate ID entered, Hunting Group set to "**Active**" FXS Port #3: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**" FXS Port #4: SIP UserID and Authenticate ID left blank, Hunting group set to "**2**" FXS Port #5: SIP UserID and Authenticate ID left blank, Hunting Group set to "**1**" FXS Port #6: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**" FXS Port #7: SIP UserID and Authenticate ID left blank, Hunting group set to "**1**" FXS Port #8: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

…

FXS Port #24: SIP UserID and Authenticate ID left blank, Hunting Group set to "**2**"

Note: A single call directed to the SIP account will NOT result in all ports ringing at the same time. They will ring in the hunting group only. This feature is applicable to incoming calls only.

There are two types of hunting groups, Linear and Circular. Linear style will sort the call to the lowest-numbered available line, this is also called "serial hunting". Circular style will distribute the calls "round- robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the

www.lapdattongdidienthoai.com

lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. These two hunting styles can be configured from the Profile_x page.

INTER-PORT CALLING

In some cases a user may want to make phone calls between the phones connected to multiple ports of the same gateway when it is used as a standalone unit, without the use of a SIP server. This feature will also be applicable when the gateway is used with Hunting Groupsand is registered to SIP server only with one master number. In such cases users still will be able to make inter-port calls by using the IVR feature.

For example, on the GXW42xx inter-port calling is achieved by dialing *** and 7 plus two extra digits corresponding to the port number. For example the user connected to port 1 can be reached by dialing *** and 701; the user connected to port 24 can be reached by dialing *** 724.

SENDING AND RECEIVING FAX

GXW42XX supports fax in two modes: 1) T.38 (Fax over IP) and 2) Fax Pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 asfax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

SUPPORT RADIUS PROTOCOL

GXW42XX supports RADIUS for authentication, authorization and billing purposes. Primary and secondary RADIUS server configurations are available to provide redundancy to this feature. In case Primary Radius server becomes unusable, RADIUS requests will be automatically sent to the secondary server. When at least one RADUIS server was configured, the device will allow users to make phone calls only after authorization from RADIUS server has been received. CDR (Call Detail Record) is also sent to the RADIUS server for billing purposes. RAIDUS server can send requests to terminate calls when run out of pre-paid credit.

The GXW42xx will be able to work in VoIP billing environment using redundant double server configuration. User will be able to configure primary and secondary RADUIS server IP Addresses or FQDNs. Once at least one RADUIS server was configured, the device will allow users to make phone calls only after permission from RADIUS server has been received. In case Primary Radius server becomes unusable, secondary will take role of primary and will manage credit recourses in the network. Imbedded RADIUS client also supports request generated by Radius server to terminate calls when run out of pre-paid credit.

CALL FEATURES

GXW42XXsupports the traditional telephony features available in a PBX as well as additional advanced telephony features.

TABLE 6: CALL FEATURES TABLE (STAR CODE)

Key	Call Features
***02**	**Forcing a Codec** (per call)*027110 (PCMU), *027111 (PCMA), *02723 (G723), *02729 (G729), *0272616 (G726-r16), *0272624 (G724-r24), *0272632 (G726-r32), *0272640 (G726-r40), *027201 (iLBC)
***03**	**Disable LEC** (pe call) Dial “*03” + ” number ”. **No** dial tone is played in the middle.
***16**	**Enable SRTP**

www.lapdattongdaidienthoai.com

***17**	**Disable SRTP**
***30**	**Block CallerID**(for all-config change)
***31**	**Send CallerID**(for all-config change)
***67**	**Block CallerID**(per call)
***82**	**Send CallerID**(per call)
***47**	**Direct IP Calling.** Dial "*47" + "IP address". **No** dial tone will be played in the middle. Detail see Direct IP Calling section on page 12.
***50**	**Disable Call Waiting** (for all-config change)
***51**	**Enable Call Waiting** (for all-config change)
***69**	**Call Return Service:** Dial *69 and the phone will dial the last incoming phone number received.
***70**	**Disable Call Waiting** (Per Call)
***71**	**Enable Call Waiting** (Per Call)
***72**	**Unconditional Call Forward:** Dial "*72" and then the forwarding number followed by "**#**". Wait for dial tone and hang up. (dial tone indicates successful forward)
***73**	**Cancel Unconditional Call Forward:** Dial "*73" and wait for dial tone, then hang up.
***74**	**Enable Paging Call:** Dial "*74" and then the destination phone number you want to activate in Paging mode.
***78**	**Enable Do Not Disturb (DND):** When enabled all incoming calls will be rejected.
***79**	**Disable Do Not Disturb (DND):** When disabled, incoming calls will be accepted.
***87**	**Blind Transfer**
***90**	**Busy Call Forward:** Dial "*90" and then the forwarding number followed by "**#**". Wait for dial tone then hang up.
***91**	**Cancel Busy Call Forward**: dial "*91". Wait for dial tone. Hang up.
***92**	**Delayed Call Forward:** Dial "*92" and then the forwarding number followed by "**#**". Wait for dial tone then hang up.
***93**	**Cancel Delayed Call Forward:** Dial "*93" for a dial tone, then hang up.
Flash/Hook	If user hears call waiting beep, flash/hook will switch to the new incoming call. Also used to switch to a new channel for a new call.
#	Pressing pound sign will serve as **Re-Dial** key.

CONFIGURATION GUIDE

CONFIGURING GXW42XX VIA VOICE PROMPT

DHCP MODE

Select voice menu option 001 to enable GXW42XXto use DHCP.

STATIC IP MODE

Select voice menu option 001 to enable GXW42XXto use STATIC IP mode, then use option 002, 003, 004, 005 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

PPPoE Mode

Select voice menu option 001 to enable GXW42XX to use PPPoEmode.

Firmware Server IP Address

Select voice menu option 013 to configure the IP address of the firmware server.

Configuration Server IP Address

Select voice menu option 014 to configure the IP address of the configuration server.

Upgrade Protocol

Select voice menu option 015 to choose firmware and configuration upgrade protocol. User can choose between TFTP, HTTP, and HTTPS.

Firmware Upgrade Mode

Select voice menu option 017 to choose firmware upgrade mode among the following three options:

1) always check, 2) check when pre/suffix changes, and 3) never upgrade

CONFIGURING GXW42XX WITH WEB BROWSER

The GXW42XXseries gateway has an embedded Web server that allows users to configure the GXW42XXthrough a web browser. It has language support to English, Chinese, French, and Spanish.

ACCESS THE WEB CONFIGURATION MENU

The GXW42XXHTML configuration menu can be accessed via Ethernet port:

To access the HTML configuration menu from the Ethernet port:

1. Follow table 4 to find the Ethernet port IP address.
2. Open a web browser, type in the IP address – for example:http://GXW42XX-IP-Address(the GXW42XXIP-Address is the Ethernet IP address for the GXW42XX).

NOTE:

- The IVR announces 12 digits IP address, you need to strip out the leading “0” in the IP address.For ex. IP address: 192.168.001.014, you need to type in http://192.168.1.14 in the web browser.

Once the HTTP request is entered and sent from a web browser, the user will see a log-in screen. There are two default passwords for the login page:

User Level:	Password:	Web pages allowed:
End User Level	123	Only Status and Basic Settings
Administrator Level	admin	Browse all pages

The password is case sensitive with maximum length of 25 characters. The factory default password for End User and administrator is “123” and “admin” respectively. Only an administrator can access the “ADVANCED SETTING”, “Profile 1~4”, and “FXS PORTs” configuration pages. Please reference the GUI pages using the following link: http://grandstream.com/products/gxw_series/gxw42xx/GUI.zip

IMPORTANT SETTINGS

The end-user must configure the following settings according to the local environment.

www.lapdationgdaidienthoai.com

NOTE: Most settings on the web configuration pages are set to the default values.

NAT SETTINGS

If you plan to keep the gateway within a *private network* behind a firewall, we recommend using STUN Server. The following three (3) settings are useful in the STUN Server scenario:

1. **STUN Server** (under Advanced Settings webpage)

   Enter a STUN Server IP (or FQDN) that you may have, or look up a free public STUN Server on the internet and enter it on this field. If using Public IP, keep this field blank.

2. **NAT Traversal** (under the Profile web pages)

   Set this to Yes when gateway is behind firewall on a private network.

DTMF METHODS

DTMF Settings are in Profile pages.

- DTMF in-audio
- DTMF via RTP (RFC2833)
- DTMF via SIP INFO

You can enable set priority of DTMF methods according to your preference, from Priority 1 to 3. This setting should be based on your server DTMF setting.

PREFERRED VOCODER (CODEC)

The GXW42XXsupports a broad range of voice codecs. Under Profile web pages, choose your preferred order of different codecs:

- PCMU/A (or G711μ/a)
- G729
- G723
- G726 (16/24/32/40)
- iLBC
- AAL2 (all G726)

This section will describe the options in the Web configuration user interface. As mentioned, a user can log in as an administrator or end-user.

Functions available for the end-user are:

- **STATUS:** Displays the network status, account status, software version and MAC-address of the phone
- **MAINTENANCE:** Basic settings such as basic network, date and time settings, and web/telnet access settings can be set here.
- **PROFILE - AUDIO SETTINGS:** DTMF, Vocoder, and Analog Line settings can be configured here for each port.

Additional functions available to administrators are:

- **MAINTENANCE:** Full settings for network, upgrade/provisioning, TR-069, Security and Syslog.
- **ADVANCED SETTINGS:** To set advanced Ring Tongs, FXO Failover, and System Features.
- **PROFILE X:** To configure each of the SIP accounts.
- **FXS PORTS:** To configure each of the FXS ports and Hunting Groups etc.

TABLE 7: STATUS

www.lapdattongdaidienthoai.com

System Info

Product Model	Contains the product model info.
Part Number	Product Part Number.
Software Version	***Program:*** This is the main software release. *This number is always used for firmware upgrade.*current version is 1.0.3.4. ***Boot:***current version is 1.0.3.2. ***Core:*** current version 1.0.3.3. ***Base:*** current version is 1.0.3.4.
System Up Time	Shows system uptime since the last reboot.
System Time	The time according to NTP server.
Service Status	Shows the status of the VOIP applications.

Network Status

MAC Address	The device ID in hexadecimal format. This is needed for Internet Service Provider troubleshooting**.** The MAC address will be used for provisioning and can be found on the label on original box and on the label located on the bottom panel of the device.
IP Address Mode	Shows the current IP mode.
IP Address	Shows IP address of GXW42XX.
Subnet Mask	ShowsSubnet Mask of GXW42XX.
Gateway	ShowsDefault Gateway of GXW42XX.
DNS Server	ShowsDNS Server of GXW42XX.
NAT Traversal	Shows type of NAT the GXW42XX is connected to via its WAN port. It is based on STUN protocol.

Port Status

Displays relevant information regarding the individual FXS ports. Example:

Port	Hook	SIP Registration	DND	Forward	Busy Forward	Delayed Forward
FXS1	On Hook	Registered	No	613		
FXS2	Off Hook	Registered	No		614	
FXS3	On Hook	Not Registered	No			
FXS4	On Hook	Registered	Yes			615
FXS…	…	…	…			

TABLE 8: MAINTENANCE

Network Settings		
IP Address Mode	Choose how the IP address obtained on the phone.	
Preferred DNS Server	Enter the preferred DNS server that should be used for DHCP and PPPoE.	
DHCP Settings	Host name (Option 12)	Specifies the name of the client. This field is optional but may be required by Internet Service Providers.
	DHCP Domain	Specifies the DHCP Domain. This value is optional, but may be required by Internet Service Providers.
	Vendor Class ID (Option 60)	Used by clients and servers to exchange vendor class ID.
Static IP Settings	Configure IP Address, Subnet Mask, Gateway, DNS Server 1, and DNS Server 2.	
Enable LLDP	Configure to enable/disable the LLDP (Link Layer Discovery Protocol) service.	
QoS Settings	Layer 3 QoS	Defines the Layer 3 QoS parameter. This value is used for IP Precedence, Diff-Serv or MPLS.
	Layer 2 QoS 802.1Q/VLAN Tag	Value used for layer 2 VLAN tag. Default setting is **blank**.
	Layer 2 QoS 802.1p Priority Value	Assigns the priority value of the Layer 2 QoS packets. Valid range is 0 to 7.
STUN Settings	Use STUN	Yes or No. Default is **No**.
	STUN server	The IP address or Domain name of the STUN server. Only non-symmetric NAT routers work with STUN.

www.lapc

	Number of STUN Response Misses Allowed	The Number of STUN response misses allowed before restarting DHCP. The minimum is 3 misses.
	Keep-Alive Interval	Specifies in seconds how often the phone sends a blank UDP packet to the SIP server in order to keep the "ping hole" on the NAT router to open. The default is 20 seconds.

Upgrade and Provisioning	
Lock Keypad Update	If set to “Yes”, the configuration update via keypad is disabled.
Firmware Upgrade and Provisioning	Specifies how firmware upgrading and provisioning request to be sent. There are three options to choose from: “Always Check for New Firmware”, “Check New Firmware only when F/W pre/suffix changes”, and “Always Skip the Firmware Check”.
XML Config File Password	The password used for encrypting the XML configuration file using OpenSSL. This is required for the phone to decrypt the encrypted XML configuration file.
HTTP/HTTPS User Name	The user name needed to authenticate withthe HTTP/HTTPS server.
HTTP/HTTPS Password	The password needed to authenticate with the HTTP/HTTPS server.
Upgrade via	Allows users to choose the firmware upgrade method via TFTP, HTTP or HTTPS.
Firmware Server Path	IP address or domain name of firmware server. That URL of the server that hosts the firmware release. The default server is: **fm.grandstream.com/gs**
Config Server Path	IP address or domain name of configuration server. The server hosts a copy of the configuration file to be installed on the gateway. The default server is: **fm.grandstream.com/gs**
Firmware File Prefix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching prefix will be downloaded.
Firmware File Postfix	This field enables user to store different versions of firmware files in one single directory on the firmware server. If configured, only the firmware file with the matching postfix will be downloaded.
Config File Prefix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching prefix will be downloaded.
Config File Postfix	This field enables user to store different configuration files in one single directory on the configuration server. If configured, only the configuration file with the matching postfix will be downloaded.

Allow DHCP Option 43 and Option 66 to Override Server	If set to "Yes", configuration and upgrade server's information can be obtained using DHCP option 66 from DHCP server. This option specifies the URL of the tftp server. **Note:**If DHCP Option 66 is enabled, the gateway will attempt downloading a configuration file from the server URL provided by DHCP, even though **Config Server Path**is left blank.
Automatic Upgrade	Choose "*Yes*" to enable automatic upgrade and provisioning. When set to No, GXW42XX will only do upgrade once at boot up. When "Check every day" or "Check every week" is checked, user can specify "Hour of the day (0-23)" or "Day of the week(0-6)". Default time is Monday 1AM.
Authenticate Conf File	If set to Yes, configuration file is authenticated before being accepted. This protects the configuration from unauthorized modifications.
Firmware Key	For firmware encryption. It should be 32-digit in Hexadecimal Representation. End user should keep it blank.

Web Port	By default, HTTP uses port 80. This field is for customizable web port.
Disable Telnet	If set to Yes, telnet access will be disabled. Default is **No**.
User Password	Set new password for web GUI access as User. This field is case sensitive with a maximum length of 30 characters.
Admin Password	Set new password for web GUI access as Admin. This field is case sensitive with a maximum length of 30 characters.
TR-069	
ACS URL	URL of the TR-069 Auto Configuration Servers (e.g., http://acs.mycompany.com, or IP address).
ACS Username	User specify the ACS Username
ACS Password	User specify the ACS password
Periodic Inform Enable	Default is **No**. If set to YES, device will send inform packets to the ACS
Periodic Inform Interval	Frequency that the inform packets will be sent out to the ACS
Connection Request Username	The user name for the TR-069 Auto Configuration Server to connect to the phone.

Connection Request Password	The password for the TR-069 Auto Configuration Server to connect to the phone.
Security Settings	
SIP TLS Certificate	The GXW42XX series supports SIP over TLS. It has built-in private key and SSL certificate. The user specified SSL certificate used for SIP over TLS is in X.509 format.
SIP TLS Private Key	You may also customize the SIP TLS Private Key. The user specified SIP TLS private key used for SIP over TLS is in X.509 format.
SIP TLS Private Key Password	SSL Private key password used for SIP Transport in TLS/TCP.
Primary RADIUS Server	Remote Authentication Dial In User Service (RADIUS). GXW42xx supports RADIUS for authentication, authorization and billing purposes. Primary and secondary RADIUS server configurations are available to provide redundancy to this feature. In case Primary Radius server becomes unusable, RADIUS requests will be automatically sent to the secondary server. When at least one RADUIS server was configured, the device will allow users to make phone calls only after authorization from RADIUS server has been received. CDR (Call Detail Record) is also sent to the RADIUS server for billing purposes. RAIDUS server can send requests to terminate calls when run out of pre-paid credit.
Primary RADIUS Authentication Port	Default is 1812. Specifies the port to be used for the Primary RADIUS Authentication.
Primary RADIUS Account Port	Default is 1813. Specifies the port to be used for the Primary RADIUS Account.
Primary RADIUS Server Secret	Specifies the secret string to be used to authenticate the RADIUS connection to the Primary Server. It should match RADIUS configuration.
Secondary RADIUS Server	Set the IP or FQDN of the Secondary RADIUS Server. In case Primary Radius server becomes unusable, secondary will take role of primary and will manage credit recourses in the network
Secondary RADIUS Authentication Port	Default is 1812. Specifies the port to be used for the Secondary RADIUS Authentication.

Secondary RADIUS Account Port	Default is 1813. Specifies the port to be used for the Secondary RADIUS Account.
Secondary RADIUS Sever Secret	Specifies the secret string to be used to authenticate the RADIUS connection to the Secondary Server. It should match RADIUS configuration.
RADIUS Timeout	Default is 2
RADIUS Retry	Default is 3
Download Device Configuration	Allows user to download and save a text file containing all the P values of each setting as configured at that point on the unit. (Note: For Security Reasons, all Passwords won't be Downloaded)
Date and Time	
NTP Server	URI or IP address of the NTP (Network Time Protocol) server. Used by the phone to synchronize the date and time. An extensive list of public NTP servers can be found at http://www.ntp.org
NTP Update Interval	Defines the update interval(in minutes) to obtain the date and time from the server.
Time Zone	Configures the date/time used on the phone according to the specified time zone.
Self-Defined Time Zone	This parameter allows the users to define their own time zone. For syntax and examples, please refer to user manual.
Syslog	
Syslog Server	The IP address or URL of System log server. The server collects system log information from the device.

Syslog Level	Select the GXW42XX to report the log level. Default is **NONE**. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: 1. product model/version on boot up (INFO level) 2. NAT related info (INFO level) 3. sent or received SIP message (DEBUG level) 4. SIP message summary (INFO level) 5. inbound and outbound calls (INFO level) 6. registration status change (INFO level) 7. negotiated codec (INFO level) 8. Ethernet link up (INFO level) 9. SLIC chip exception (WARNING and ERROR levels) 10. memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message *Example*: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up
Print SIP in Syslog	Enable or disable printing of full SIP messages in Syslog.

TABLE 9: ADVANCED SETTINGS

Ring Tones	
System Ring Cadence	Configuration option for all FXS ports ring cadence for all incoming calls. (Syntax: c=on1/off1-on2/off2-on3/off3; [...]) Default is set to c=2000/4000; (US standards)

www.vattonic

Call Progress Tones	Using these settings, user can configure tone frequencies according to user preference. By default, the tones are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ONis the period of ringing (ON time in ms) while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. • "Dial tone" • "Ringback tone" • "Busy tone" • "Reorder tone" • "Confirmation tone" • "Call-Waiting tone" Please refer the document below to determine your local call progress tones: http://www.itu.int/ITU-T/inr/forms/files/tones-0203.pdf
FXO Failover	
Failover to FXO Gateway	Enable or disable the FailoverFXO Gateway.
FXO Gateway IP	IP Address or URI of the FXO gateway.
System Features	
Disable Direct IP Call	Disables the Direct IP Call function. Default is "**No**". If set to "**Yes**" direct IP-to-IP calling will not be supported.
Disable SIP NOTIFY Authentication	Disables challenging SIP NOTIFY reboot and resync messages.
Disable Voice Prompt	Disables the voice prompt configuration. Default is "**No**". If set to "**Yes**" accessing integrated voice menu will be impossible.
IVR Language	Choose English, Chinese, or Spanish.
Display Language	Choose language for LCD display.
Prompt Dial Tone Code	Simulates an analog PBX where a code is required to dial an outside line.

TABLE 10: PROFILES

General Settings	

Profile Active	When set to Yes, the SIP Profile is activated.
SIP Server	SIP Server's IP address or Domain name provided by VoIP service provider.
Failover SIP Server	Failover SIP Server's IP address or Domain name provided by VoIP Service provider. This server will be used if the Primary SIP server becomes unavailable.
Prefer Primary SIP Server	Default is No. If set to yes it will register to Primary Server if registration with Failover server expires
Outbound Proxy	IP address or Domain name of Outbound Proxy, or Media Gateway, or Session Border Controller. Used by GXW42XX for firewall or NAT penetration in different network environments. If symmetric NAT is detected, STUN will not work and ONLY outbound proxy can correct the problem.
Network Settings	
DNS Mode	One from the 3 modes available for "DNS Mode" configuration: -A Record (for resolving IP Address of target according to domain name) -SRV (DNS SRV resource records indicates how to find services for various protocols) -NAPTR/SRV (Naming Authority Pointer according to RFC 2915) -Use Configured IP (If selected, please fill in Primary IP, Backup IP 1 and Backup IP 2.) One mode can be chosen for the client to look up server. The default value is "A Record"
Primary IP	Configures the primary IP address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.
Backup IP 1	Configures the backup IP 1 address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.
Backup IP 2	Configures the backup IP 2 address where the phone sends DNS query to when "Use Configured IP" is selected for DNS mode.

NAT Traversal	This parameter defines whether the GXW42XX NAT traversal mechanism is activated or not. If activated (by choosing "STUN") and a STUN server is also specified, then the GXW42XX performs according to the STUN client specification. Under this mode, the embedded STUN client will detect if and what type of firewall/NAT is being used. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the GXW42XX will use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "STUN" with no specified STUN server, the GXW42XX will periodically (every 20 seconds) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Use NAT IP	The NAT IP address used in SIP/SDP messages. It should ONLY be used if required by your ITSP.
Proxy-Require	A SIP Extension to notify the SIP server that the phone is behind a NAT/Firewall.

SIP Settings - Basic Settings

SIP transport	User can select UDP or TCP or TLS. Please make sure you're SIP Server or network environment supports SIP over the selected transport method. Default is UDP.
SIP Registration	This parameter controls whether the GXW42XX needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is **No**. If set to "Yes", the SIP user's registration information is cleared on reboot.
Add Auth Header on Initial REGISTER	Default is **No**. If set to "Yes", an Authentication Header with blank nonce will be added in the initial REGISTER.
Outgoing Calls Without Registration	Default is **No**. If set to "Yes," user can place outgoing calls even when not registered (if allowed by Internet Telephone Service Provider) but is unable to receive incoming calls. Any port, member of a Hunting Group that is not registered with a SIP account, will be able to place outbound calls using the SIP credentials of the primary Hunting Group port. For example:Port 1, 3 and 5 are members of the same Hunting Group. Port 1 is registered with a SIP account. Ports 3 and 5 are not registered. Ports 3 and 5 will be able to place outbound calls using the SIP account of port 1, even if ***Outgoing Call without Registration***is set to No
Register Expiration	Allows the user to specify the time frequency (in minutes) for the GXW42XX to refresh its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).

SIP Registration Failure Retry Wait Time	Allows the user to specify the time frequency (in seconds) for the GXW42XX to re-register after registration failure. The default interval is 20 seconds. The maximum interval is 3600seconds (1 hour).
Reregister Before Expiration	Determines how many seconds before the previous registration expires that the port should reregister.
Local SIP Port	Defines the local SIP port the GXW42XX will listen and transmit. The default value for Profile 1 is 5060 and 6060 for Profile 2.
Use Random SIP Port	Default is **No**. If set to Yes, the device will pick randomly-generated SIP ports. This is usually necessary when multiple GXW42XX/HT50X are behind the same NAT.
SIP T1 Timeout	T1 is an estimate of the round-trip time between the client and server transactions. If the network latency is high, select larger value for more reliable usage.
SIP T2 Timeout	Maximum retransmission interval for non-INVITE requests and INVITE responses.
Remove OBP from Route Header	Default is **No**. If set to Yes, the Outbound Proxy will be removed from the route header.
Support SIP Instance ID	Default is **Yes**. If set to Yes, the contact header in REGISTER request will contain SIP Instance ID as defined in IETF SIP Outbound draft.
Refer-To Use Target Contact	Default is **No**. If set to YES, then for Attended Transfer, the "Refer-To" header uses the transferred target's Contact header information.
SUBSCRIBE for MWI	Default is **No**. When set to Yes, a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Enable 100rel	Enables the use of PRACK (Provisional Acknowledgment) method.
TEL URI	The default setting is "Disabled". If the phone has an assigned PSTN Number, this field should be set to "User=Phone" then a "User=Phone" parameter will be attached to the "From header" in the SIP request to indicate the E.164 number. If server supports TEL URI format, then this option needs to be selected.
Use Request Routing ID in SIP Headers	Default is **No**. If set to Yes, device will use the configured [Request URI Routing ID] in the SIP Header. This option is usually used under a SIP trunk account's configuration.
SIP Settings - Session Timer	

Session Expiration	Grandstream implemented SIP Session Timer. The session timer extension enables SIP sessions to be periodically “refreshed” via a SIP request (UPDATE, or re-INVITE. When the session interval expires, if there is no refresh via a UPDATE or re-INVITE message, the session will be terminated. Session Expiration is the time (in seconds) at which the session is considered timed out, if no successful session refresh transaction occurs beforehand. The default value is **180 seconds**.
Min-SE	The minimum session expiration (in seconds). The default value is **90 seconds**.
Caller Request Timer	If selecting “Yes” the phone will use session timer when it makes outbound calls if remote party supports session timer.
Callee Request Timer	If selecting “Yes” the phone will use session timer when it receives inbound calls with session timer request.
Force Timer	If selecting “Yes” the phone will use session timer even if the remote party does not support this feature. Selecting “No” will allow the phone to enable session timer only when the remote party support this feature. To turn off Session Timer, select “No” for Caller Request Timer, Callee Request Timer, and Force Timer.
UAC Specify Refresher	As a Caller, select UAC to use the phone as the refresher, or UAS to use the Callee or proxy server as the refresher.
UAS Specify Refresher	As a Callee, select UAC to use caller or proxy server as the refresher, or UAS to use the phone as the refresher.
Force INVITE	Session Timer can be refreshed using INVITE method or UPDATE method. Select “Yes” to use INVITE method to refresh the session timer.
SIP Settings - Security Settings	
Validate Incoming Messages	Defines whether the incoming messages will be validated.
Check SIP User ID for Incoming INVITE	If set to "Yes", SIP User ID will be checked in the Request URI of the incoming INVITE. If it doesn't match the phone's SIP User ID, the call will be rejected. Direct IP calling will also be disabled.
Accept Incoming SIP from Proxy Only	If set to "Yes", SIP User ID will be checked in the Request URI of the incoming INVITE. If it doesn't match the phone's SIP User ID, the call will be rejected. Direct IP calling will also be disabled.

Authenticate Incoming INVITE	If set to "Yes", the phone will challenge the incoming INVITE for authentication with SIP 401 Unauthorized response.
Fax Settings	
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)
Fax Tone Detection Mode	Default is **Callee**.This decides whether Caller or Callee sends out the re-INVITE for T.38 or Fax Pass Through.
Send Re-INVITE After Fax Completion	Default is **No**, If set to "Yes", device will send an INVITE with audio vocoders upon completition of Fax to continue session in audio only.
Send Re-INVITE After Fax Tone	If set to "Yes", device will send an Re-INVITE after Fax tone is detected, disabling will only work under Broadsoft feature.
Enable Silence Detection for Fax Disconnect	For fax machines that do not send a Disconnect when fax is done. This option Enables/Disables the detection of silence in order to know the fax has finished. The silence period is non-configurable and fixed to 7 seconds.
Audio Settings	
Preferred DTMF method (in listed order)	The GXW42XX supports up to 3 different DTMF methods including in-audio, via RTP (RFC2833) and via Sip Info. The user can configure DTMF method in a priority list.
Disable DTMF Negotiation	Default is **No**. If set to yes,use above DTMF order without negotiation
DTMF Payload Type	Sets the payload type for DTMF using RFC2833.
Preferred Vocoder	The GXW42XX supports up to 5 different Vocoder types including G.711 A-/U-law, G.726 (Supports bit rates 16, 24, 32 and 40), G.723.1, G.729A/B, iLBC. The user can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder is entered by choosing the appropriate option in "Choice 1". The last Vocoder is entered by choosing the appropriate option in "Choice 8".
Voice Frames per TX	Number of the frame size when it transmits. Default is **2**, from 1 -4 for G711/G726/G729
G723 Rate	Defines the encoding rate for G.723 vocoder. By default, 6.3kbps rate is chosen.
G726-32 Packing Mode	Choose the packing mode for G726-32

iLBC Frame Size	Sets the iLBC frame size in 20ms or 30ms
iLBC Payload type	Default value is **97**.Defines payload type for iLBC. The valid range is between 96 and 127.
AAL2-G726-16 Payload type	Default value is**100**. Range is from 96 to 127.
AAL2-G726-24 Payload type	Default value is **99**. Range is from 96 to 127.
AAL2-G726-32 Payload type	Default value is **104**. Range is from 96 to 127.
AAL2-G726-40 Payload type	Default value is **103**. Range is from 96 to 127.
Use First Matching Vocoder in 200OK SDP	Default is **No**. If set to "Yes", device will include only the first match vocoder in its 200OK response, otherwise it will include all match vocoders in same order received in INVITE.
SRTP Mode	Default is **Disabled**. Other options are **Enabled but not forced**, and **Enabled and forced**. It uses SDP Security Description to exchange key.Please refer SDES: http://www.apps.ietf.org/rfc/rfc4568.html SRTP: http://www.apps.ietf.org/rfc/rfc3711.html
Silence Suppression (VAD)	Default is **No**.VAD allows detecting the absence of audio and conserve bandwidth by preventing the transmission of "silent packets" over the network.
Jitter Buffer Type	Select either Fixed or Adaptive based on network conditions.
Jitter Buffer Length	Select Low, Medium or High based on network conditions. • High (initial 200ms, min 40ms, max 600ms) Note: not all vocoders can meet the high requirement • Medium (initial 100ms, min 20ms, max 200ms) • Low (initial 50ms, min 10ms, max 100ms)
SLIC Setting	Depends on standard phone type (and location)
Caller ID Scheme	Select the value according to the local Telco standard where the GXW42XX is installed. Please refer to the pull down list to select.
Polarity Reversal	Default is **No**. If set to "Yes", polarity will be reversed upon call establishment and termination.

Loop Current Disconnect	Set to Yes if the traditional PBX you are using with GXW42XX uses this method for signaling call termination. Default is **No**.
Loop Current Disconnect duration	Default is **200**.In 100 - 10000 milliseconds range.
Enable Hook Flash	If no, flash will be treated as an on hook event.
Hook Flash timing(Minimum)/ (Maximum)	Sets the minimum/maximum time the phone is on hook before being detected as a hook flash. The range is 40 to 2000 milliseconds
On Hook Timing	Sets the time required to detect that the phone is on hook. The range is 40 to 2000 milliseconds. Default is 400 milliseconds.
Gain	Handset volume adjustment. RX is for receiving volume(direction FXS→to analog phone), TX is for transmission volume(Analog phone→ to FXS). Default values are **0dB** for both parameters. Loudest volume: +6dB Lowest volume: -6dB.
Disable Line Echo Canceller (LEC)	Default is **No**. If set to "Yes", device will not use LEC to remove echo from a voice communication.
SLIC Setting	Depends on standard phone type (and location)
Caller ID Scheme	Select the value according to the local Telco standard where the GXW42XX is installed. Please refer to the pull down list to select.
Call Settings	
Early Dial	Default is **No**. Use only if proxy supports 484 response. This parameter controls whether the phone will send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "Re-Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will likely be rejected by the proxy (with a 404 Not Found error). *This feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling*.
Dial Plan Prefix	Sets the prefix added to each dialed number.

Dial Plan	**Dial Plan Rules**: 1. Accept Digits: 1,2,3,4,5,6,7,8,9,0 , *, #, A,a,B,b,C,c,D,d 2. Grammar: x - any digit from 0-9; a. xx+ - at least 2 digits number; b. xx – at least 2 digits number; c. ^ - exclude; d. [3-5] - any digit of 3, 4, or 5; e. [147] - any digit 1, 4, or 7; f. <2=011> - replace digit 2 with 011 when dialing g. < =1> - add a leading 1 to all numbers dialed, vice versa will remove a 1 from the number dialed h. | - or • **Example 1:** {[369]11 | 1617xxxxxxx} – Allow 311, 611, 911, and any 10 digit numbers of leading digits 1617 • **Example 2:** {^1900x+ | <=1617>xxxxxxx} – Block any number of leading digits 1900 and add prefix 1617 for any dialed 7 digit numbers • **Example 3:** {1xxx[2-9]xxxxxx | <2=011>x+} – Allow any length of number with leading digit 2 and 10 digit-numbers of leading digit 1 and leading exchange number between 2 and 9; If leading digit is 2, replace leading digit 2 with 011 before dialing. 3. Default: Outgoing - {x+} Example of a simple dial plan used in a Home/Office in the US: **{ ^1900x. | <=1617>[2-9]xxxxxx | 1[2-9]xx[2-9]xxxxxx | 011[2-9]x. | [3469]11 }** ***Explanation*** of example rule (reading from left to right): • **^1900x. -** prevents dialing any number started with 1900 • **<=1617>[2-9]xxxxxx -** allows dialing to local area code (617) numbers by dialing 7 numbers and 1617 area code will be added automatically • **1[2-9]xx[2-9]xxxxxx |-** allows dialing to any US/Canada Number with 11 digits length • **011[2-9]x. -** allows international calls starting with 011 **[3469]11 -** allow dialing special and emergency numbers 311, 411, 611 and 911 **Note:** In some cases user wishes todial strings such as *123 to activate voice mail or other application provided by service provider. In this case * should be predefined inside dial plan feature and the Dial Plan will be: **{ [x*]+}**.

Use # as Dial Key	Allows users to configure the "#" key as the "Send" (or "Dial") key. If set to "Yes", "#" will send the number. In this case, this key is essentially equivalent to the "Dial" key. If set to "No", this "#" key can be included as part of number.
No Key Entry Timeout	Default is **4 seconds**. Call will be completed within this time interval if no additional key entry occurs.
Off-Hook Auto-Dial Delay	Determines how many seconds after off-hook to wait before autodialing the extension set under Advanced Port Settings. The range is 0 to 60 seconds.
Enable Call Features	Default is **Yes**. (If Yes, call features using star codes will be supported locally)
Disable Call Waiting	Default is **No**. If set to **YES**Call Waiting indication information will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Caller ID	Default is **No**. If set to **YES** Call Waiting caller ID will not be provided to analog phone connected to this FXS port.
Disable Call Waiting Tone	Default is **No**. This is to disable the stutter Call Waiting Tone when a Call Waiting call arrives. The CWCID will still be displayed.
Disable Receiver Offhook Tone	Default is **No**. If **YES**, ROH tone will not be played after off hook for 60 seconds.
Disable Reminder Ring for On-Hold Call	Default is **No**. This is to disable the Reminder Ring that is played when a call is waiting on hold and the analog phone goes on-hook.
Disable Visual MWI	Visual message indicator is a special on-hook caller ID type message that enables and disables the message waiting light on certain phones. GXW42XX has this feature enabled by default. However, certain phones (rare) that do not support it may mistakenly treat this CID signal as an incoming call. A configuration option is needed to turn on MWI in this case.
Visual MWI Type	This is the type of signal sent to the analog phone to make it turn the lamp ON upon receiving a Voice mail. Check the phone's manual to find out what signal is supported, FSK (default) or Neon.
Transfer on Conference Hangup	Defines whether the call is transferred to the other party if the conference initiator hangs up.
Disable Bellcore-style 3-Way Conference	Disable Bellcore-style 3-Way Conference

Send Hook Flash Event	Default is **No**. If set to yes, flash will be sent as a DTMF event.
Ring Timeout	Incoming call will stop ringing when not picked up given a specific period of time.
Delayed Call Forward Wait Timeout	Default value is **20 seconds**. In case this feature activated using * codes (*92 code), the call will be forwarded after this preconfigured amount of time.
Send Anonymous	If this parameter is set to "Yes", the "From" header along with Privacy and P_Asserted_Identity headers in outgoing INVITE message will be set to anonymous, blocking Caller ID.
Anonymous Call Rejection	Default is **No**. If set to Yes, incoming calls with anonymous Caller ID will be rejected with 486 Busy message.
Hunting Group Type	Linear and Circular. Linear style will sort the call to the lowest-numbered available line, this is also called "serial hunting". Circular style will distribute the calls "round-robin". If a call is assigned to line 1, the next call goes to 2 and the next to 3. The succession throughout each of the lines continues even if one of the previous lines becomes available. When the end of the hunt group is reached, the hunting starts over at the first line. Lines are skipped if they are still busy on a previous call. These two hunting styles can be configured from the Profile_x page.
Hunting Group Ring Timeout	Default is **20 seconds**. If call is not answered within this designated time period, the call will be forwarded to the next member of a Hunt Group.
Special Feature	Default is **Standard**. Choose the selection to meet some special requirements from Softswitch vendors. Example of vendors - CBCOM, RNK.

Ring Tones

Distinctive Ringtone	Custom Ring Tone 1 to 3 with associate Caller ID: when selected, if Caller ID is configured, then the device will ONLY uses this ring tone when the incoming call is from the Caller ID. System Ring Tone is used for all other calls. When selected but no Caller ID is configured, the selected ring tone will be used for **all** incoming calls.Distinctive ring tones can be configured not only for matching whole number, but also for matching prefixes. In this case symbol * (star) will be used. If server supports Alert-Info header and standard ring tone set (Bellcore) or distinctive ring tone 1-10 is specified, then the ring tone in the Alert-Info header from server will be used. **For example:** If configured as *617, Ring Tone 1 will be used in case of call arrived from Massachusetts. Any other incoming call will ring using cadence defined in parameter System Ring Cadence located under Advanced Settings Configuration page.

Ring Tones	Configure ring cadences according to preference.

TABLE 11: FXS PORTS

Port Settings	SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
	Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User ID.
	Password	SIP service subscriber's account password for GXW42XX to register to (SIP) servers of ITSP.
	Name	Any name to identify this specific user.
	Profile	Select the corresponding Profile ID (1/2/3/4)
	Enable FXS (TR-069)	Enable or disable the FXS port.
Advanced Port Settings	Offhook Auto-dial	Configure an auto dial number when offhook.
	Hunting Group	Assign this port to a certain hunting group
	Request URI Routing ID	If configured, device will route the incoming call to designated port by request URI user ID in SIP INVITE.
FXO Mapping	Map to FXO Port #	This is used only when peering with a Grandstream GXW410x. Default 1, Supported values are 1-8, meaning line 1 to 8 of the GXW410x device where the port will be mapped to.
	Map to FXO Gateway IP	This is used when peering with an FXO gateway of any brand. You have to specifically mention the IP and sip port where call will be sent to.
	And Port	Sip port that will be annexed to the IP address above.

SAVING THE CONFIGURATION CHANGES

After user makes a change to the configuration, press the "Update" button in the Configuration Menu. Theweb browser will then display a message window to confirm saved changes. Grandstream recommends reboot or power cycle the gateway after saving changes.

REBOOTING FROM REMOTE

Press the "Reboot" button at the bottom of the configuration menu to reboot the phone remotely. The web browser will then display a message window to confirm that reboot is underway. Wait 30 seconds to log in again.

CONFIGURATION THROUGH A CENTRAL SERVER

Grandstream GXW42xx can be automatically configured from a central provisioning system.

When GXW42xx boots up, it will send TFTP or HTTP/HTTPS requests to download configuration files, “cfg000b82xxxxxx” and “cfg00082xxxxxx.xml”, where “000b82xxxxxx” is the LAN MAC address of the GXW42xx. If the download of “cfgxxxxxxxxxxxx.xml” is not successful, the provision program will issue request a generic configuration file “cfg.xml”. Configuration file name should be in lower case letters.

The configuration data can be downloaded via TFTP or HTTP/HTTPS from the central server. A service provider or an enterprise with large deployment of GXW42xx can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a central provisioning system GAPS (Grandstream Automated Provisioning System) to support automated configuration of Grandstream devices. GAPS uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual Grandstream device for firmware upgrade, remote reboot, etc.

Grandstream provides GAPS service to VoIP service providers. Use GAPS for either simple redirection or with certain special provisioning settings. At boot-up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer’s TFTP or HTTP/HTTPS server for further provisioning.

Grandstream also provides configuration tools (Windows and Linux/Unix version) to facilitate the task of generating device configuration files. The Grandstream configuration tools are free to end users. The configuration tools and configuration templates are available for download from http://www.grandstream.com/support/tools .

SOFTWARE UPGRADE

Software upgrade can be done via either TFTP or HTTP/HTTPS. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

FIRMWARE UPGRADE THROUGH TFTP/HTTP/HTTPS

To upgrade via TFTP or HTTP/HTTPS, the “Firmware Upgrade and Provisioning upgrade via” field needs to be set to TFTP HTTP or HTTPS, respectively. “Firmware Server Path” needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

e.g.firmware.mycompany.com:6688/Grandstream/1.0.4.4

e.g.firmware.grandstream.com

NOTES:

- Firmware upgrade server in IP address format can be configured via IVR. Please refer to theCONFIGURATION GUIDE section for instructions. If the server is in FQDN format, it must be set via the web configuration interface.

www.laptoptongdaidienhoai.com

- Grandstream recommends end-user use the Grandstream HTTP server. Its address can be found athttp://www.grandstream.com/support/firmware . Currently the HTTP firmware server URL is firmware.grandstream.com. For large companies, we recommend to maintain their own TFTP/ HTTP/HTTPS server for upgrade and provisioning procedures.
- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the GXW will attempt to retrieve the new image files by downloading them into the GXW420x 's SRAM. During this stage, the GXW's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP/HTTPS fails for any reason (e.g. TFTP/HTTP/HTTPS server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the GXW will stop the TFTP/HTTP/HTTPS process and simply boot using the existing code image in the flash.
- Firmware upgrade may take as long as 15 to 30 minutes over Internet, or just 5 minutes if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade.
- Grandstream's latest firmware is available http://www.grandstream.com/support/firmware.

  Oversea users are strongly recommended to download the binary files and upgrade firmware locally in a controlled LAN environment.
- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://www.solarwinds.com/products/freetools/free_tftp_server.aspx. Our latest official release can be downloaded from http://www.grandstream.com/y-firmware.htm.

INSTRUCTIONS FOR LOCAL FIRMWARE UPGRADE:

1. Unzip the file and put all of them under the root directory of the TFTP server.
2. Put the PC running the TFTP server and the GXW42XXdevice in the same LAN segment.
3. TFTP server's security settings should be changed from "Receive Only" to "Transmit Only" for the firmware upgrade.
4. Configure the Firmware Server Path with the IP address of the PC
5. Update the change and reboot the unit.

CONFIGURATION FILE DOWNLOAD

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP/HTTPS. "Config Server Path" is the TFTP or HTTP/HTTPS server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be same or different from the "Firmware Server Path".

www.lapdatongdaidienthoai.com

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 1 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When a Grandstream device boots up or reboots, it will issue a request for a configuration file "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". In addition, device will also requests a XML configuration file "cfgxxxxxxxxxxxx.xml". If the download of "cfgxxxxxxxxxxxx.xml" is not successful, the provision program will issue a request for a generic configuration file "cfg.xml". Configuration file name should be in lower case letters.

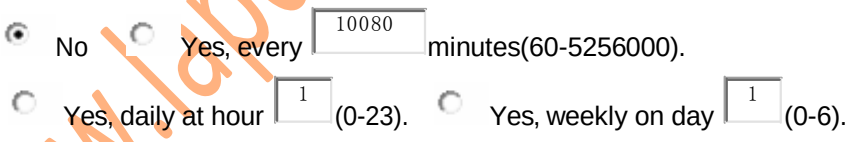
FIRMWARE AND CONFIGURATION FILE PREFIX AND POSTFIX

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it possible to store ALL of the firmwares with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.
In addition, whenthe field "Check New Firmware only when F/W pre/suffix changes" is selected, the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

MANAGING FIRMWARE AND CONFIGURATION FILE DOWNLOAD

When "Automatic Upgrade" is set to "Yes", Service Provider can use P193 to have the devices periodically check with either Firmware Server or Config Server, whenever they are defined. This allows the device periodically check whether there is any new changes need to be taken, similar to the AntiVirus Software to upgrade the Virus Definition files. Screenshot is below:

Automatic Upgrade:



www.lapdattongdaidienthoai.com

RESTORE FACTORY DEFAULT SETTING

FACTORY RESET

WARNING!Restoring the Factory Default Setting will DELETE all configuration information of the phone. Please BACKUP or PRINT out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your VoIP service provider.

There are two (2) methods for resetting your unit:

RESET BUTTON

Reset default factory settings following these four (4) steps:

1. Unplug the Ethernet cable.
2. Locate a needle-sized hole on the back panel of the gateway unit next to the power connection.
3. Insert a pin in this hole, and press for more than 4 seconds.
4. Take out the pin. All unit settings are restored to factory settings.

IVR COMMAND

Reset default factory settings using the IVR Prompt (Table 5):

1. Dial "***" for voice prompt.
2. Enter "099" and wait for "reset" voice prompt.
3. Enter the encoded MAC address (Look below on how to encode MAC address).
4. Wait 15 seconds and device will automatically reboot and restore factory settings.

Encode the MAC Address

1. Locate the MAC address of the device. It is the 12 digit HEX number on the bottom of the unit.
2. Key in the MAC address. Use the following mapping:

   0-9: 0-9

   A: 22 (press the "2" key twice, "A" will show on the LCD)

   B: 222

   C: 2222

   D: 33 (press the "3" key twice, "D" will show on the LCD)

   E: 333

   F: 3333

For example: if the MAC address is 000***b***8200***e***395, it should be keyed in as "000***222***8200***333***395".

NOTE:

1. Factory Reset will be disabled if the "**Lock keypad update**" is set to "Yes".
2. If still have difficulties to get access to the device, pushing the reset button during booting up will trigger the **RECOVERY MODE**, and device will use static IP 192.168.1.234/24. Put your PC under the

www.lapdattongdaidienthoai.com

same subnet, and then you will be able to open the RECOVERY MODE web page on 192.168.1.234 by your web browser.

www.lapdattongdaidienthoai.com